

Understanding MP3 Audio Compression

by Robert Kelley

This article will show what an MP3 file looks like, how it works, and a general description of how to decode one.

Overview

The MP3 audio format has become somewhat of a standard method for deploying video and audio media across the Internet. It is used for both live streaming of Internet broadcasts and for storing media files to be downloaded and played back on a PC. However, its most well-known use is in reproducing CD-quality music for distribution over the Internet. There is a fair amount of software available to accomplish this, but not a lot of documentation to explain how it is done. This article will show what an MP3 file looks like, how it works, and a general description of how to decode one.

MP3 is short for the MPEG-x Layer 3 compression algorithm, where x is either '1' or '2.' MPEG stands for Motion Pictures Experts Group. The name refers to a group of developers that work under the joint direction of the International Standards Organization (ISO) and the International Electro-Technical Commission (IEC). Together, these organizations are responsible for the continued evolution of the MPEG algorithms as a whole.

The MPEG format is currently split into two flavors: MPEG-1 and

MPEG-2. The first was completed in November 1992, while the latter in November 1994. MPEG-2 adds several new audio features to MPEG-1. These new audio features are a "low sample rate extension," to address very low bit rate applications with limited bandwidth requirements, and a "multi-channel extension," to address surround-sound applications with up to five main audio channels. This extension

also provides for an optional extra "low frequency enhancement (LFE)" channel for subwoofer signals. In addition to this, a "multilingual extension" allows the inclusion of up to seven more audio channels.

Each of these two formats is composed of three layers: Layer 1, Layer 2, and Layer 3. These layers differ in that they provide progressively better compression, with increased encoder/decoder com-

plexity and delays.

Backward compatibility between the three layers has been maintained by keeping the same file format. However, due to the differences in encoding, the layers are not upward compatible.

For instance, a Layer 3 decoder

Sound Quality	Bandwidth	Mode	Bit Rate	Reduction Ratio
Telephone sound	2.5 kHz	Mono	8 kbps	96:1
Better than short-wave	4.5 kHz	Mono	16 kbps	48:1
Better than AM radio	7.5 kHz	Mono	32 kbps	24:1
Similar to FM radio	11 kHz	Stereo	56...64 kbps	26...24:1
Near-CD	15 kHz	Stereo	96 kbps	16:1
CD	>15 kHz	Stereo	112...128 kbps	14...12:1

FIGURE 1: MPEG LAYER 3 TYPICAL PERFORMANCE DATA

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The IJ-90V uses external memory and comes in either blue, red, yellow, black, silver, or purple. Memory card readers are available for both the PC and Macintosh, including JS-100C Compact Flash Memory cards with parallel port, JS-101C with USB ports, and PCA-100C for portable computers with PCMCIA slots. It has a suggested retail price of \$179.00.

can decode any Layer 1 or Layer 2 encoded sample, but a Layer 2 decoder cannot decode a Layer 3 encoded sample. This article deals with the third layer, which is the most widely used and is what is employed in the "MP3" format.

In terms of compression, MPEG-2 Layer 3 encoding can achieve CD-quality compression at 12:1. Figure 1 shows the various compression rates that can be achieved, and the associated signal quality that comes with them.

How Does It Work

MPEG, in any layer, is a perceptual coding scheme. This means that it exploits the inherent properties of the human ear while trying to maintain the original sound quality of the sample. These properties appear as limitations in that the ear has trouble identifying more than one frequency if a tone becomes dominant for a specific period of time.

The frequencies in the vicinity of this tone are less audible, if the energy falls below a certain threshold called a masking threshold. In

the most simple of terms, the algorithm removes the inaudible portions of the sample leaving only what the human ear can hear. The compression results in very little signal degradation, but the original sample can never be recovered. Instead, the signal will be perceptually identical to the human ear.

The MP3 Format

The content of the MP3 file is organized into a number of frames. The frame consists of information to construct 1,152 samples of data. The samples are sub-divided into two granules, each 576 samples in size. Each granule contains Scale Factors and Huffman Encoded Bits, which is where the actual audio data is located, and will be discussed in further detail later. The composition of the frame takes on five distinct parts: a Header, CRC, Side Information, Main data, and Ancillary data. Figure 2 illustrates this.

Header

The Header portion of the MP3

frame is a fixed 32 bits in size and contains a synchronization word and system information. The first 12 bits of the header are known as the synchronization word. This word marks the beginning of the frame and must be set to the binary equivalent of all ones. See Figure 3 for a graphical breakdown of the header. The following is a detailed breakdown of each element of the header, and what it means:

ID: The ID of the audio algorithm used. A '1' indicates that ISO/IEC 11172-3 audio was used.

Layer: Indicates which layer was used. Possible values are 1, 2, or 3. Zero is not defined.

Protection Bit: A value of '1' indicates that there are two CRC bytes appended to the end of the header.

Bit Rate Index: Indicates the bit rate used. Allowable bit rates vary depending upon which layer is used. The index corresponds to an index into an array of possible bit

rate values.

Sampling Frequency: Indicates the sample frequency of the audio sample. This value is also an index into an array of possible values.

Padding Bit: Indicates whether or not an additional byte has been added to the frame in order to adjust the mean bit rate to the sample frequency.

Private Bit: A bit designated for private use.

Mode: One of four possible mode values: Stereo, Joint-Stereo, Dual Channel, or Single Channel.

Mode Extension: Indicates which type of joint stereo method was used: Intensity Stereo or MS Stereo.

Copyright: Indicates if a copyright exists.

Original/Copy: A '1' indicates that the sample is an original, a '0' indicates that it is a copy.

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Emphasis: Indicates what type of pre-emphasis must be used.

CRC

The CRC is two bytes in length and is optionally found after the header. It provides a limited amount of error checking primarily for the decoder. Because only the most critical portion of the frame is included in the CRC, it's composed of only bits 16 through 31 in both the header portion, and the side information portion.

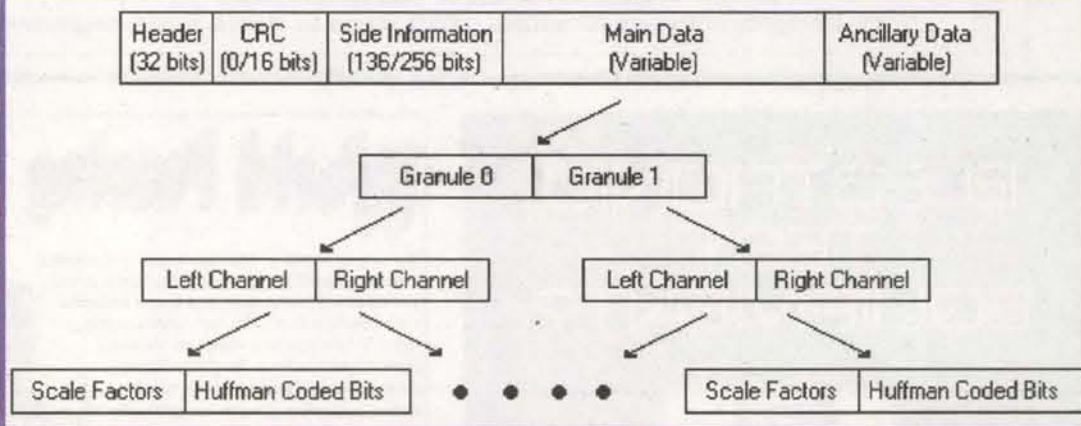
An error occurring in any other part of the frame will corrupt the data itself, and may or may not be distinguishable during playback. However, an error in these critical bits will corrupt the frame and the results will be much more noticeable.

Standard error handling procedure is to mute the bad frame or retransmit it. Ultimately, it is up to the specific implementation of the decoder.

Side Information

The primary purpose of the side information section of the frame is to hold the necessary information to

Figure 2: MPEG Layer 3 Frame Organization. This diagram represents a dual-channel frame (stereo). Note: The Side Information section of the frame has a similar organization except that it contains information common to each channel in each granule.



decode the main data. Without it, the main data cannot be decoded properly, hence the need for the optional CRC implementation to verify this information.

Specifically, this section contains Huffman tables, which are used during the Huffman decoding process (explained later) and information for the reconstruction of the scale factors. Both of these elements of data are located in the Main Data portion of the frame.

The side information also indicates where the main data begins in the current frame via a nine-bit main data pointer. This is necessary because of the use of a mechanism known as a "bit reservoir." The bit

reservoir is used to hold extra data in the next frame should the main data require more space than is available in the current frame.

This section is 17 bytes in length for single channel mode, and 32 bytes for dual channel mode. See Figure 4 for a breakdown of Side Information. The following is a detailed breakdown of each element of Side Information, and what it means:

Main Data Begin: A nine-bit pointer to the location of the start of the Main Data that belongs to the given frame. This pointer value is a negative offset from the first byte of the synchronization word.

Private Bits: These bits are user-defined and can be used by the encoder.

SCFSI: (Scale Factor Selection Information) Determines whether one scale factor is sent for each granule, or for both granules, per channel. Four bits are required for this per channel, and each bit corresponds to a group of scale factor bands. A value of zero means that scale factors are transmitted for each granule for that particular group of scale factor bands. A one means that the same scale factors are used, therefore only the scale factors for the first granule are transmitted.

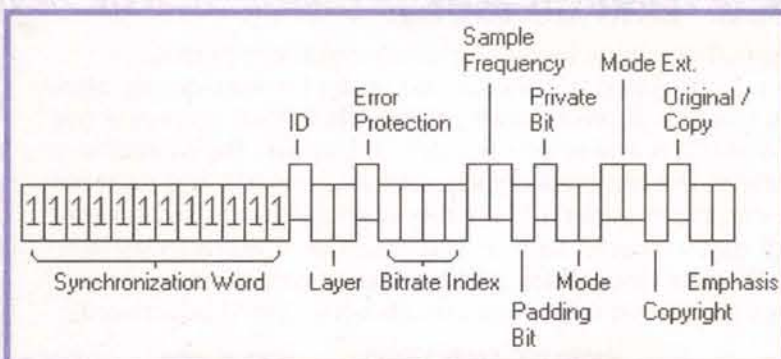


Figure 3:
MPEG
Layer 3.
Header
Format.
Note:
Each box
represents
one bit.

Side Information for each granule

The side information specific to each granule is broken out in Figure 5. The following is an explanation of each of the elements:

Par23Length: The number of main data bits allocated for Scale Factors and Huffman Encoded Data. This variable can be used to determine the beginning of the next granule since the width of Side Information is constant.

Big Values: Indicates the size of the big value portion of each granule. The Big Value portion of the granule is defined as one of three partitions of values after the quantization of the 576 samples of the granule. The partitions include

rzero, count1, and big values.

Global Gain: Contains information about the step size used in the quantizer where the quantization is done logarithmically.

Scale Factor Compress: Variable transmitted for each granule. Indicates the number of bits used for the transmission of scale factors. It is an index into a table of values that returns two numbers which correspond to the number of bits in the first and second group of scale factor bands.

Windows Switching Flag: A flag that indicates whether or not a normal window is being used. A value of zero defines the normal window; a non-zero value indicates some other type of window. This flag influences what the next 22/44

bits of data will look like.

Block Type: Indicates what type of window is used in a particular granule.

Mixed Block Flag: A flag that indicates that different types of windows is used in the lower and higher frequencies. If set, it means that the two

lowest sub-bands are transformed using a normal window (as if Block Type was set to zero), and the rest of the bands are transformed using whatever window is specified in Block Type.

Table Select: Allows for the use of different Huffman code tables. There are 32 possible tables that can be used.

Sub-block Gain: Enables a gain by a factor of four for one sub-block. This feature only works when short windows are chosen (Block Type = 2).

Region 0 Count: In order to improve the performance of the Huffman encoder, the Big Values portion of the Granule Side Information is divided into three regions. This is the size of region 0.

Region 1 Count: The size (number of bits), of region 1 of the Big Values data.

Pre-flag: Indicates that additional amplification was used in the higher frequencies of the quantized values. The value of this variable is added to the scale factors and is not available when short windows are chosen (Block Type = 2).

Scale Factor Scale: Determines how the scale factors are quantized logarithmically. The possible methods are with a step size of two, or the square root of two.

Count 1 Table Select: A flag that indicates which of two possible Huffman code tables was used to code the partition count1.

Main Data

Main Data consists of three types of data: Scale Factors, Huffman coded bits, and Ancillary Data.*

Scale Factors: The Scale Factors are used to color the quantization noise. They are transmitted for every group of frequency lines, or Scale Factor Bands. Scale factors can be sent for each scale factor band per granule or not. This is determined by the before-mentioned SCFSI variable. The exact number sent also depends upon what type of Window has been specified.

Huffman Coded Bits: The Huffman Coded bits are one of the major reasons why the MP3 format can achieve such high compression rates while still maintaining superior sample quality. They work via the encoding of the likeness of data within the stream. A table is first created showing the distribution of like and unlike data across the stream. This table is then parsed and codes are assigned to each entry. Data that has a high probability of being found more often are assigned a short code, while data that is rarely present are assigned a much longer code. Thus, decoding the data results in a new representation of it, but with fewer bits.

Ancillary Data

Ancillary data is optional data added to the end of the file. It is most commonly used to provide information about the media encoded, such as the name of a song. The size of this data is variable, and is determined by subtracting the sum of the header, CRC, and audio data from the total size of the frame. It lies from the end of the Huffman encoded bits to the loca-

Main Data Begin	Private Bits	SCFSI	Side Info Gr. 0	Side Info Gr. 1
9 bits	5 or 3 bits	4 or 8 bits	59 or 118 bits	59 or 118 bits

Figure 4: Format of Side Information.
Note: Elements with two bit values reflect mono or stereo mode, respectively.

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tion pointed to by the next Main Data pointer.

Decoding the Data

Once a frame has been received, the first thing done must be to parse the header. First, the synchronization bits must be verified. If they are all set, then the synchronization is valid and the CRC can be checked, if it is present. If the CRC checks out, then it is time to collect data from the Side Information section of the frame to learn the characteristics of the Huffman Encoded bits. This includes where to find the Huffman Encoded bits in the stream, which tables to use to decode them, and whether ESCAPE codes are present. In addition, the decoder must ensure that all 576 frequency lines have been generated even though not all of them may have been used. In that case, the unused lines must be padded with zeroes.

Now that it is known how the Huffman Encoded bits were encoded, they can be decoded. The end result of running the algorithm is to obtain the scaled, quantized frequency lines. These lines are then used with the results of the Scale Factor Decoding algorithm to begin the De-scaling process.

The Scale Factor Decoding algorithm decodes the scale factors, which are the first part of the Main Data section of the frame. Both the Scale Factor info and the coded scale factors are used as input to this algorithm, and are found in the Side Information section of the frame.

It is during the next phase of decoding known as De-scaling, where the real benefits of MP3 compression are realized. As mentioned earlier, the scaled and quantized frequency lines generated by the Huffman Decoder are used together with the reconstructed scale factors from the Scale Factor Decoding algorithm to generate a perceptual copy of the original frequency lines.

The next step in decoding the data is to perform Reordering. Reordering is necessary because the frequency lines generated by the De-scaling algorithm are not always ordered in the same way. This occurs when the encoder uses different window methods. A long window would generate frequency lines ordered first by sub-band, then by frequency. With short windows, the reverse is true. Therefore, this must detect what type of window method was used, and reorder the values, if necessary.

At this point, all of the frequency lines are properly ordered, and primarily decoded. Now it is time to

Par23Length 12 or 24 bits Block Type	Big Values 9 or 18 bits Mixed Block Flag	Global Gain 8 or 16 bits Table Select	Scale Factor Compress 4 or 8 bits Sub-block Gain	Windows Switching Flag 1 or 2 bits
2 or 4 bits Table Select	1 or 2 bits Region 0 Count	10 or 20 bits Region 1 Count	9 or 18 bits	
15 or 30 bits Pre-flag	4 or 8 bits Scale Factor Scale	3 or 6 bits Count 1 Table Select		
1 or 2 bits	1 or 2 bits	1 or 2 bits		

Figure 5: Format of Side Information for each granule (1 and 2) section of the Side Information block. Note: The data bits in the first row are always transmitted. The data bits in only one of the second or third rows are transmitted, depending upon the value of Windows Switching Flag. The fourth row of data bits is always appended to either the second or third.

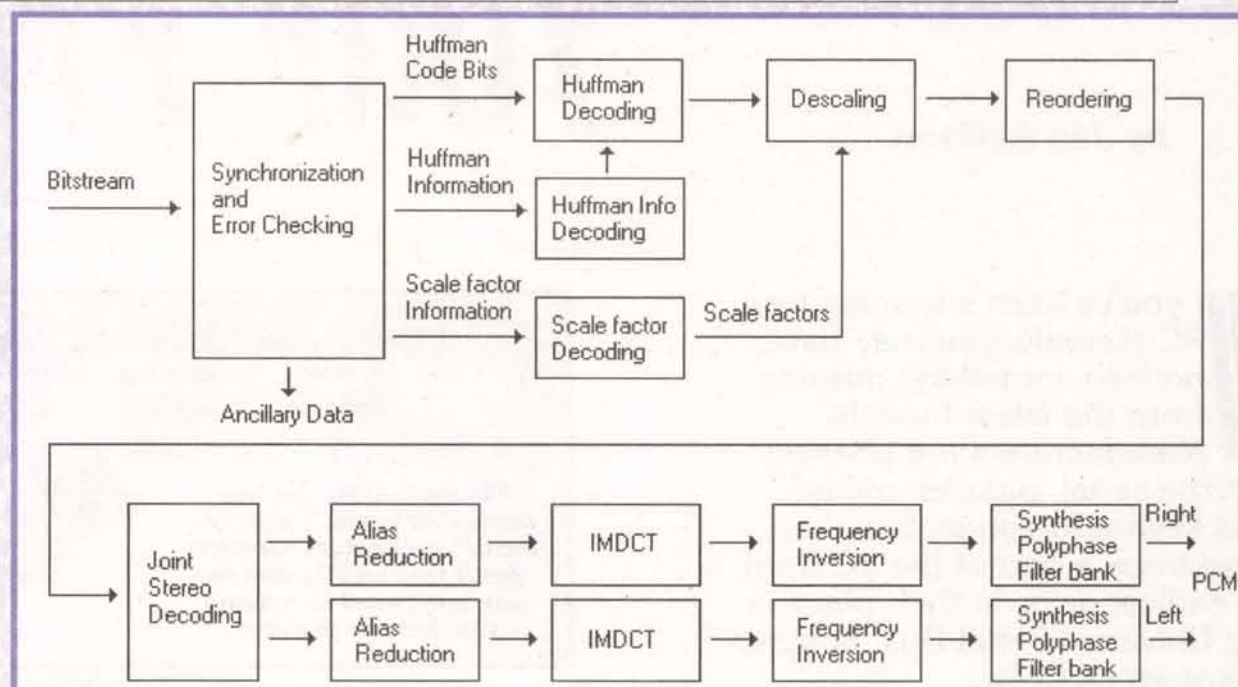


Figure 6: Decoder flow diagram. Shows the steps that an MP3 decoder must take to convert a compressed sample back into PCM (Pulse Code Modulated) signal format.

perform the necessary processing to convert the encoded stereo signal, if it is a stereo sample, into separate left/right stereo signals. From this point on, the rest of the algorithms applied to the data will do so for each of the two separated stereo signals. The following algorithms are a bit more complicated, and will be touched on briefly.

The first of these algorithms is called Alias Reduction. In this phase, the aliasing artifacts are added to the signal gain. This process consists of eight butterfly calculations for each sub-band.

Next, an Inverse Modified Discrete Cosine Transformation (IMDCT) is performed. The frequency lines from the alias reduction are mapped to polyphase filter sub-band samples. This calculation differs between different values of Block Type. Possible frequency inversions in the polyphase filter bank, created from the previous algorithm, are handled next by multiplying every odd time sample of every odd sub-band by -1. This is known as the Frequency Inversion step.

The last step is to synthesize the polyphase filter bank. Each time a sample, from each of the 32 sub-bands has been calculated, it is

applied to the synthesis polyphase filter bank and 32 consecutive audio samples are calculated.

After running this algorithm on the frequency lines, the end result of the decode is achieved with the data in raw, PCM (Pulse Code Modulated) format. The data is packed in one or two bytes per sample, depending upon which mode is being employed, mono or stereo, respectively. This raw data can be streamed directly to a sound card and played back. Windows Wave (.WAV) files often use this format, which is why files of this type tend to be so large. Figure 6 shows the entire decode process from beginning to end.

Summary

This article only scratches the surface on providing the amount of information necessary in order to create an MP3 decoder. Unfortunately for media player developers, information about the MP3 file format is difficult to come by.

Only a handful of sources contributed to the information contained within this article as it is expensive to purchase the specifica-

tions directly from IEEE. Source code is available on the Internet, which is written in 'C' and/or Assembly, that applies this information to the development of an MP3 encoder/decoder.

The following FTP site contains some public domain source code written in the C programming language. This code was written for demonstration purposes, but it should prove a sufficient springboard for creating an MP3 Encoder/Decoder. The site is: [ftp.iis.fhg.de/pub/layer3/public_c/](ftp.iis.fhg.de/pub/layer3/public_c/).
NV

Contributing work

1. The Masters Thesis of Kent Salomonsen, Sten Sogaard, and Eddie Proft Larsen for the Institute of Electronic Systems, Department of Communication Technology, at Aalborg University, Denmark.

2. Fraunhofer IIS web site: www.iis.fhg.de